WHAT IS CLAIMED IS:

1. A method of generating a wide-band speech signal from a first narrow-band speech signal, the method comprising:

analyzing the first narrow-band speech signal to generate one or more parameters;

synthesizing a first higher frequency-band signal based on at least one of the one or more parameters;

generating a second higher frequency-band signal by amplifying the first higher-frequency band signal by a gain amount that is based, at least in part, on one or more spectral amplitude peaks in the first narrow-band speech signal; and

combining the second higher frequency-band signal with a second narrow-band speech signal that is derived from the first narrow-band speech signal.

2. The method of claim 1, further comprising generating the second narrow-band speech signal by a technique that includes up-sampling the narrow-band speech signal.

3. The method of claim 1, wherein analyzing the first narrow-band speech signal to generate one or more parameters comprises using linear prediction to generate an error signal from the first narrow-band speech signal.

4. The method of claim 1, wherein:

the one or more parameters include signal spectrum information that identifies harmonic tones of the narrow-band speech signal; and

generating the first higher frequency-band signal based on at least one of the one or more parameters comprises generating a spectrally copied signal that has a signal spectrum in a higher frequency region that replicates the harmonic tones of the narrow-band speech signal during voiced speech segments.

5. The method of claim 4, wherein generating the first higher frequency-band signal further comprises generating a bandpass filtered signal by bandpass filtering the spectrally copied signal.

6. The method of claim 5, wherein generating the first higher frequency-band signal further comprises formant filtering the bandpass filtered signal.

7. The method of claim 4, wherein generating the first higher frequency-band signal further comprises:

generating a bandpass filtered signal by bandpass filtering the spectrally copied signal; and

formant filtering the bandpass filtered signal only if the narrow-band speech signal is judged to represent voiced speech.

8. The method of claim 4, wherein generating the first higher frequency-band signal further comprises formant filtering the spectrally copied signal.

9. The method of claim 1, wherein:

the one or more parameters include a set of amplitude parameters that are proportional to amplitudes of pole frequency components of the first narrow-band speech signal; and

amplifying the first higher frequency-band signal comprises:

using a first gain amount if the first narrow-band speech signal is judged to represent voiced speech; and

using a second gain amount if the first narrow-band speech signal is judged to represent fricated speech.

10. The method of claim 9, wherein amplifying the first higher frequency-band signal further comprises:

using a third gain amount if the first narrow-band speech signal is judged to represent neither voiced nor fricated speech.

11. The method of claim 10, wherein the third gain amount is a very low constant gain amount.

12. The method of claim 9, wherein:

the amplitude parameters are logarithmically scaled;

using the first gain amount comprises making a first linear combination of the amplitude parameters; and

using the second gain amount comprises making a second linear combination of the amplitude parameters.

13. The method of claim 1, wherein the second narrow-band speech signal is the first narrow-band speech signal.

14. The method of claim 1, further comprising:

synthesizing a lower frequency-band signal based on at least one of the one or more parameters, and

wherein combining the second higher frequency-band signal with a second narrow-band speech signal that is derived from the first narrow-band speech signal comprises combining the second higher frequency-band signal, the second narrow-band speech signal that is derived from the first narrow-band speech signal and the lower frequency-band signal.

15. The method of claim 14, wherein:

the one or more parameters include a pitch frequency parameter; and

synthesizing the lower frequency-band signal based on at least one of the one or more parameters comprises generating continuous sine tones that are based on the pitch frequency parameter.

16. The method of claim 15, wherein:

the narrow-band speech signal comprises a plurality of narrow-band speech signal segments;

the pitch frequency parameter is estimated for each of the narrow-band speech signal segments; and

the continuous sine tones are changed gradually during a first part of each speech signal segment.

17. The method of claim 16, wherein synthesizing the lower frequency-band signal based on at least one of the one or more parameters further comprises adaptively changing an amplitude level of the continuous sine tones based on an amplitude level of at least one formant in the narrow-band speech signal segment.

18. The method of claim 17, wherein the at least one formant in the narrow-band speech signal segment is a first formant in the narrow-band speech signal segment.

19. The method of claim 17, wherein adaptively changing the amplitude level of the continuous sine tones based on the amplitude level of at least one formant in the narrow-band speech signal segment comprises:

adaptively changing an amplitude level of the continuous sine tones by an amount, $g_l(m)$, given by:

$$g_l(m) = C_l \cdot \sqrt{\frac{\sum_{l=0}^{p} a(l)\cdot\gamma_{xx}(l)}{\left|\sum_{l=0}^{p} a(l)\cdot e^{-j2\pi l f_{Nl}}\right|^2}} \quad ,$$

where C_l is a constant; m is a segment number; γ_{xx} is an autocorrelation value of the narrow-band speech signal, x ; f_{Nl} is a frequency of a first formant of the narrow-band speech signal; and p is an order of a linear prediction filter.

20. The method of claim 17, wherein the continuous sine tones, $s(n)$, are generated in accordance with:

$$s(n)=\sum_{i=1}^{N} s_i(n),$$

where the summation range $i=1$ to N is selected such that all sine tones will be added together, and:

$$s_i(n) = \begin{cases} \left(g_l(m-1)+n\frac{g_l(m)-g_l(m-1)}{L_l}\right)\sin\left(i(\phi(m)+n)\left(\omega(m-1)+n\frac{\omega(m)-\omega(m-1)}{L_l}\right)\right), & n=0,\ldots,L_l \\ g_l(m)\sin(i(\phi(m)+n)\omega(m)), & n=L_l+1,\ldots,L-1 \end{cases}$$

where $\phi(m)$ is a phase compensation needed to maintain a continuous sinusoid within segments, $\omega(m)$ is the pitch frequency of a current speech signal segment m, L is the number of samples in each speech signal segment, and L_l is the end sample of the soft transition within each speech signal segment.

21. The method of claim 15, wherein synthesizing the lower frequency-band signal based on at least one of the one or more parameters further comprises lowpass filtering the continuous sine tones.

22. The method of claim 21, wherein lowpass filtering the continuous sine tones is performed with an upper cutoff frequency substantially equal to 300 Hz.

23. An apparatus for generating a wide-band speech signal from a first narrow-band speech signal, the method comprising:

logic that analyzes the first narrow-band speech signal to generate one or more parameters;

logic that synthesizes a first higher frequency-band signal based on at least one of the one or more parameters;

logic that generates a second higher frequency-band signal by amplifying the first higher-frequency band signal by a gain amount that is based, at least in part, on one or more spectral amplitude peaks in the first narrow-band speech signal; and

logic that combines the second higher frequency-band signal with a second narrow-band speech signal that is derived from the first narrow-band speech signal.

24. The apparatus of claim 23, further comprising logic that generates the second narrow-band speech signal by a technique that includes up-sampling the narrow-band speech signal.

25. The apparatus of claim 23, wherein the logic that analyzes the first narrow-band speech signal to generate one or more parameters comprises logic that uses linear prediction to generate an error signal from the first narrow-band speech signal.

26. The apparatus of claim 23, wherein:

the one or more parameters include signal spectrum information that identifies harmonic tones of the narrow-band speech signal; and

the logic that generates the first higher frequency-band signal based on at least one of the one or more parameters comprises logic that generates a spectrally copied signal that has a signal spectrum in a higher frequency region that replicates the harmonic tones of the narrow-band speech signal during voiced speech segments.

27. The apparatus of claim 26, wherein the logic that generates the first higher frequency-band signal further comprises logic that generates a bandpass filtered signal by bandpass filtering the spectrally copied signal.

28. The apparatus of claim 27, wherein the logic that generates the first higher frequency-band signal further comprises a formant filter that formant filters the bandpass filtered signal.

29. The apparatus of claim 26, wherein the logic that generates the first higher frequency-band signal further comprises:

a bandpass filter that generates a bandpass filtered signal by bandpass filtering the spectrally copied signal; and

a formant filter that formant filters the bandpass filtered signal only if the narrow-band speech signal is judged to represent voiced speech.

30. The method of claim 26, wherein the logic that generates the first higher frequency-band signal further comprises a formant filter that formant filters the spectrally copied signal.

31. The apparatus of claim 23, wherein:

the one or more parameters include a set of amplitude parameters that are proportional to amplitudes of pole frequency components of the first narrow-band speech signal; and

the logic that amplifies the first higher frequency-band signal comprises:

logic that uses a first gain amount if the first narrow-band speech signal is judged to represent voiced speech; and

logic that uses a second gain amount if the first narrow-band speech signal is judged to represent fricated speech.

32. The apparatus of claim 31, wherein the logic that amplifies the first higher frequency-band signal further comprises:

logic that uses a third gain amount if the first narrow-band speech signal is judged to represent neither voiced nor fricated speech.

33. The apparatus of claim 32, wherein the third gain amount is a very low constant gain amount.

34. The apparatus of claim 31, wherein:

the amplitude parameters are logarithmically scaled;

the logic that uses the first gain amount comprises logic that makes a first linear combination of the amplitude parameters; and

the logic that uses the second gain amount comprises logic that makes a second linear combination of the amplitude parameters.

35. The apparatus of claim 23, wherein the second narrow-band speech signal is the first narrow-band speech signal.

36. The apparatus of claim 23, further comprising:

logic that synthesizes a lower frequency-band signal based on at least one of the one or more parameters, and

wherein the logic that combines the second higher frequency-band signal with a second narrow-band speech signal that is derived from the first narrow-band

speech signal comprises logic that combines the second higher frequency-band signal, the second narrow-band speech signal that is derived from the first narrow-band speech signal and the lower frequency-band signal.

37. The apparatus of claim 36, wherein:

the one or more parameters include a pitch frequency parameter; and

the logic that synthesizes the lower frequency-band signal based on at least one of the one or more parameters comprises logic that generates continuous sine tones that are based on the pitch frequency parameter.

38. The apparatus of claim 37, wherein:

the narrow-band speech signal comprises a plurality of narrow-band speech signal segments;

the pitch frequency parameter is estimated for each of the narrow-band speech signal segments; and

the continuous sine tones are changed gradually during a first part of each speech signal segment.

39. The apparatus of claim 38, wherein the logic that synthesizes the lower frequency-band signal based on at least one of the one or more parameters further comprises logic that adaptively changes an amplitude level of the continuous sine tones based on an amplitude level of at least one formant in the narrow-band speech signal segment.

40. The apparatus of claim 39, wherein the at least one formant in the narrow-band speech signal segment is a first formant in the narrow-band speech signal segment.

41. The apparatus of claim 39, wherein the logic that adaptively changes the amplitude level of the continuous sine tones based on the amplitude level of at least one formant in the narrow-band speech signal segment comprises:

logic that adaptively changes an amplitude level of the continuous sine tones by an amount, $g_l(m)$, given by:

$$g_l(m)=C_l \cdot \sqrt{\frac{\sum_{l=0}^{p} a(l)\cdot\gamma_{xx}(l)}{\left|\sum_{l=0}^{p} a(l)\cdot e^{-j2\pi l f_{Nl}}\right|^2}} ,$$

where C_l is a constant; m is a segment number; γ_{xx} is an autocorrelation value of the narrow-band speech signal, x ; f_{Nl} is a frequency of a first formant of the narrow-band speech signal; and p is an order of a linear prediction filter.

42. The apparatus of claim 39, wherein the continuous sine tones, $s(n)$, are generated in accordance with:

$$s(n)=\sum_{i=1}^{N} s_i(n),$$

where the summation range $i=1$ to N is selected such that all sine tones will be added together, and:

$$s_i(n) = \begin{cases} \left(g_l(m-1)+n\frac{g_l(m)-g_l(m-1)}{L_l}\right)\sin\left(i(\phi(m)+n)\left(\omega(m-1)+n\frac{\omega(m)-\omega(m-1)}{L_l}\right)\right), & n=0,\dots,L_l \\ g_l(m)\sin(i(\phi(m)+n)\omega(m)), & n=L_l+1,\dots,L-1 \end{cases}$$

where $\phi(m)$ is a phase compensation needed to maintain a continuous sinusoid within segments, $\omega(m)$ is the pitch frequency of a current speech signal segment m, L is the number of samples in each speech signal segment, and L_l is the end sample of the soft transition within each speech signal segment.

43. The apparatus of claim 37, wherein the logic that synthesizes the lower frequency-band signal based on at least one of the one or more parameters further comprises a lowpass filter that lowpass filters the continuous sine tones.

44. The apparatus of claim 43, wherein the lowpass filter has an upper cutoff frequency substantially equal to 300 Hz.